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(54) DISTRIBUTED CODEC FOR PACKET-BASED COMMUNICATIONS

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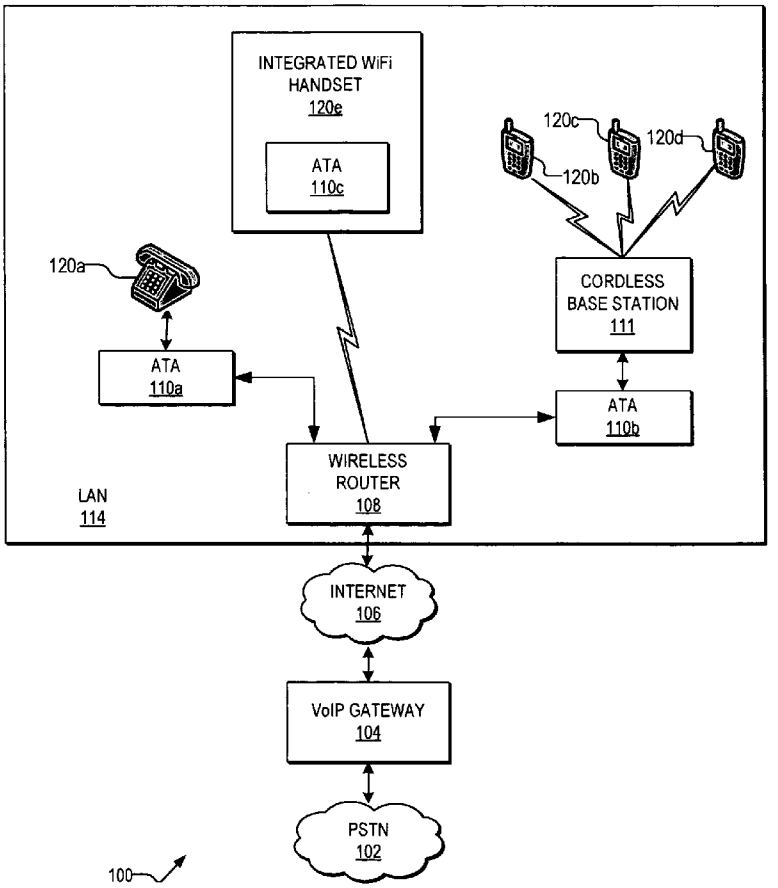
(57) **ABSTRACT**

A packet-based communications system is provided, including: a client communications module for transmitting digital audio signals to a user interface device over a local area network (LAN) and for receiving digital audio signals from the user interface device, a voice packet module, and a network protocol module for transmitting data packets between the voice packet module and an Internet Protocol (IP) network. The voice packet module is configured to: receive audio signals from the client communications module and encapsulate the audio signals into data packets; convert data packets into audio signals and transmit the audio signals to the client communications module. A communications device is provided, including: a microphone; a speaker; a user input device; a network interface for coupling the communications device to a local area network (LAN); an audio processing module comprising an analog-to-digital converter (ADC) for converting analog audio signals from the microphone into a digital audio signal and a digital-to-analog converter (DAC) for converting a digital audio signal into an analog audio signal, said analog signal being output by the speaker; and an endpoint communications module for connecting to a telephony adapter on the LAN via the network interface, said endpoint communications module communicating digital audio signals with the telephony adapter over the LAN.



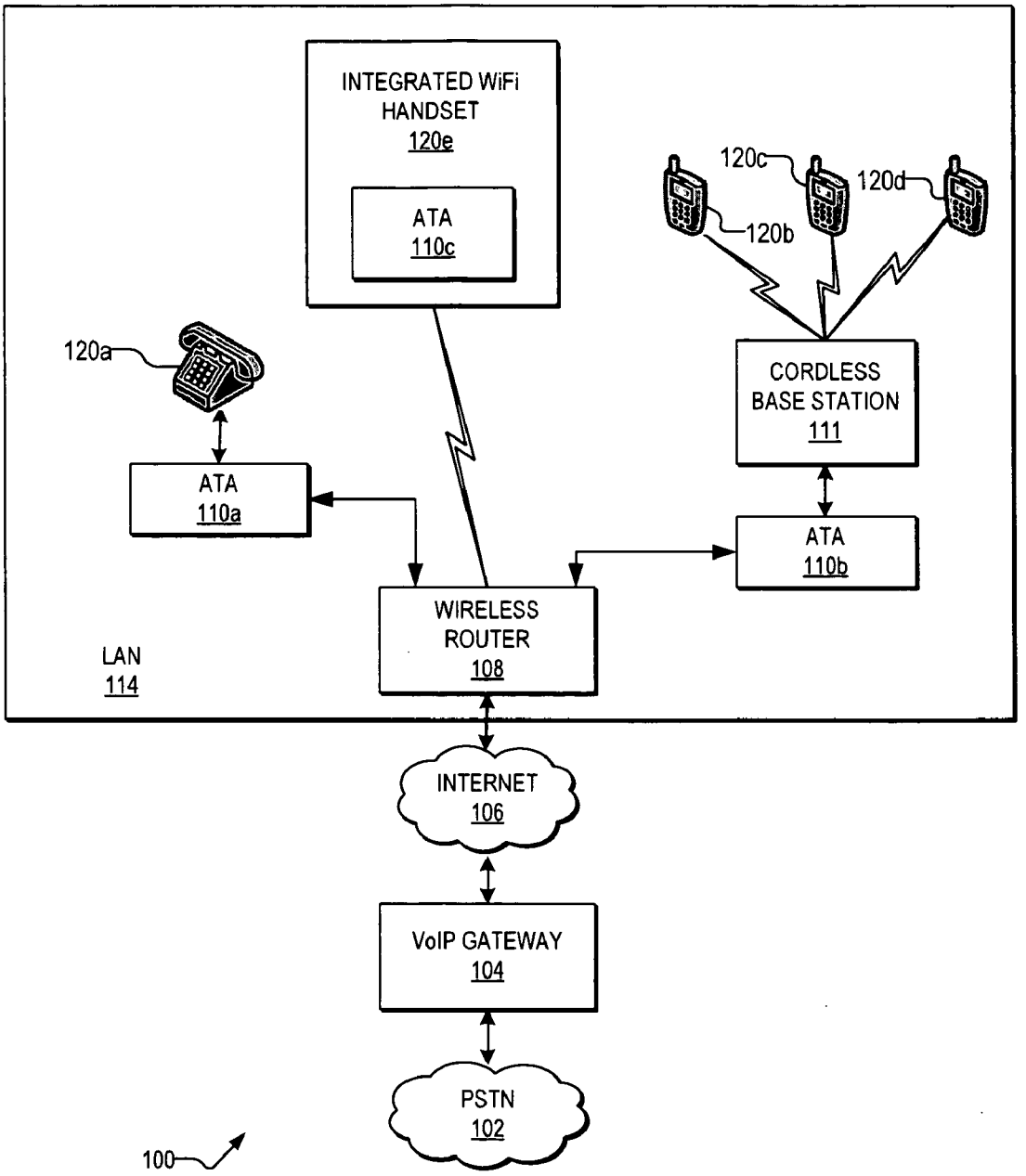

INTEGRATED WiFi HANDSET
120e
ATA
110c
120c
120d
120b
120a
CORDLESS BASE STATION
111
ATA
110a
ATA
110b
WIRELESS ROUTER
108
LAN
114
INTERNET
106
VoIP GATEWAY
104
PSTN
102
100

FIG. 1

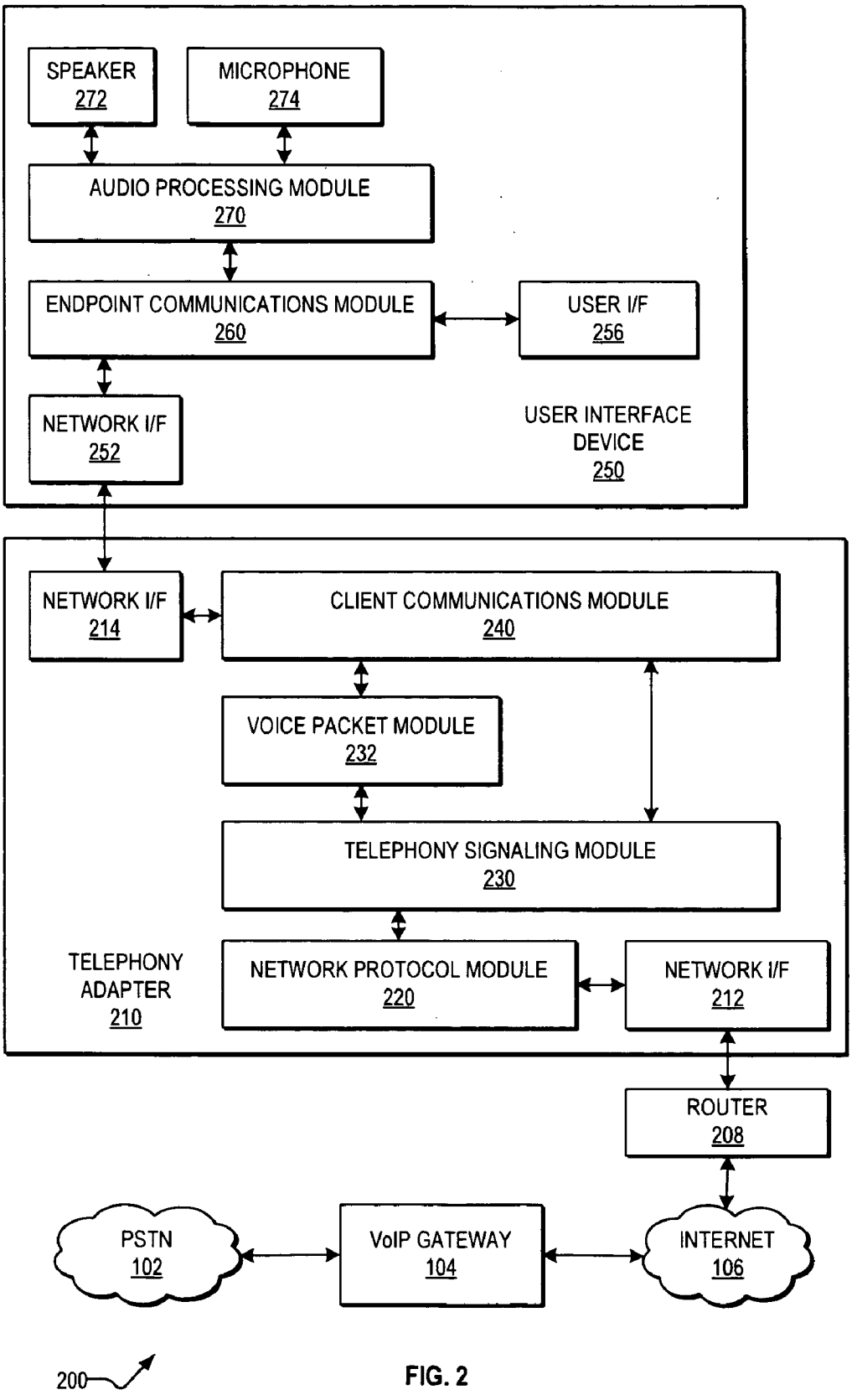

SPEAKER
272
MICROPHONE
274
AUDIO PROCESSING MODULE
270
ENDPOINT COMMUNICATIONS MODULE
260
USER I/F
256
NETWORK I/F
252
USER INTERFACE
DEVICE
250
NETWORK I/F
214
CLIENT COMMUNICATIONS MODULE
240
VOICE PACKET MODULE
232
TELEPHONY SIGNALING MODULE
230
TELEPHONY
ADAPTER
210
NETWORK PROTOCOL MODULE
220
NETWORK I/F
212
ROUTER
208
PSTN
102
VoIP GATEWAY
104
INTERNET
106
200

FIG. 2

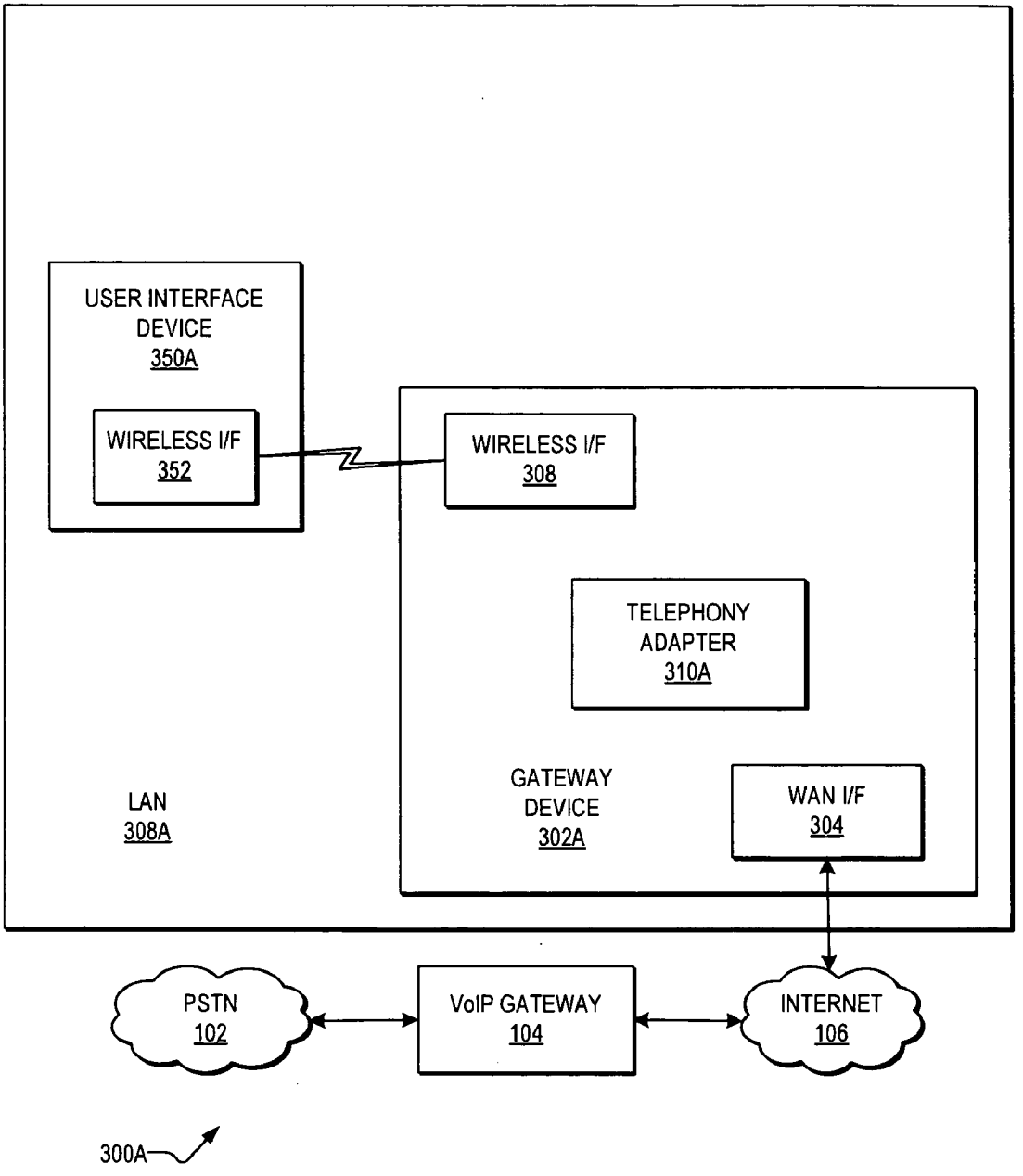

USER INTERFACE DEVICE
350A
WIRELESS I/F
352
WIRELESS I/F
308
TELEPHONY ADAPTER
310A
GATEWAY DEVICE
302A
WAN I/F
304
LAN
308A
PSTN
102
VoIP GATEWAY
104
INTERNET
106
300A

FIG. 3A

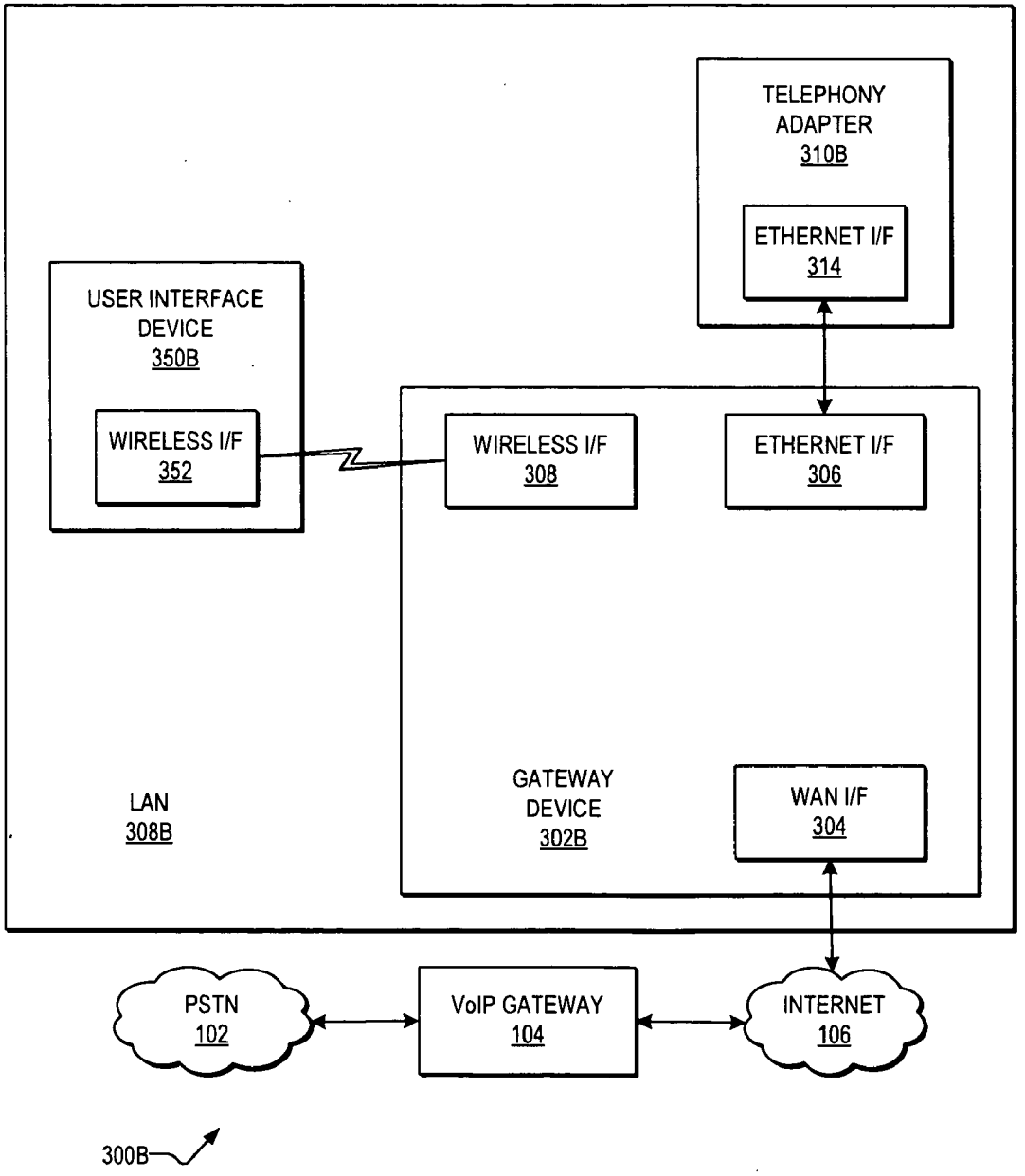

TELEPHONY ADAPTER 310B
ETHERNET I/F 314
USER INTERFACE DEVICE 350B
WIRELESS I/F 352
WIRELESS I/F 308
ETHERNET I/F 306
LAN 308B
GATEWAY DEVICE 302B
WAN I/F 304
PSTN 102
VoIP GATEWAY 104
INTERNET 106
300B

FIG. 3B

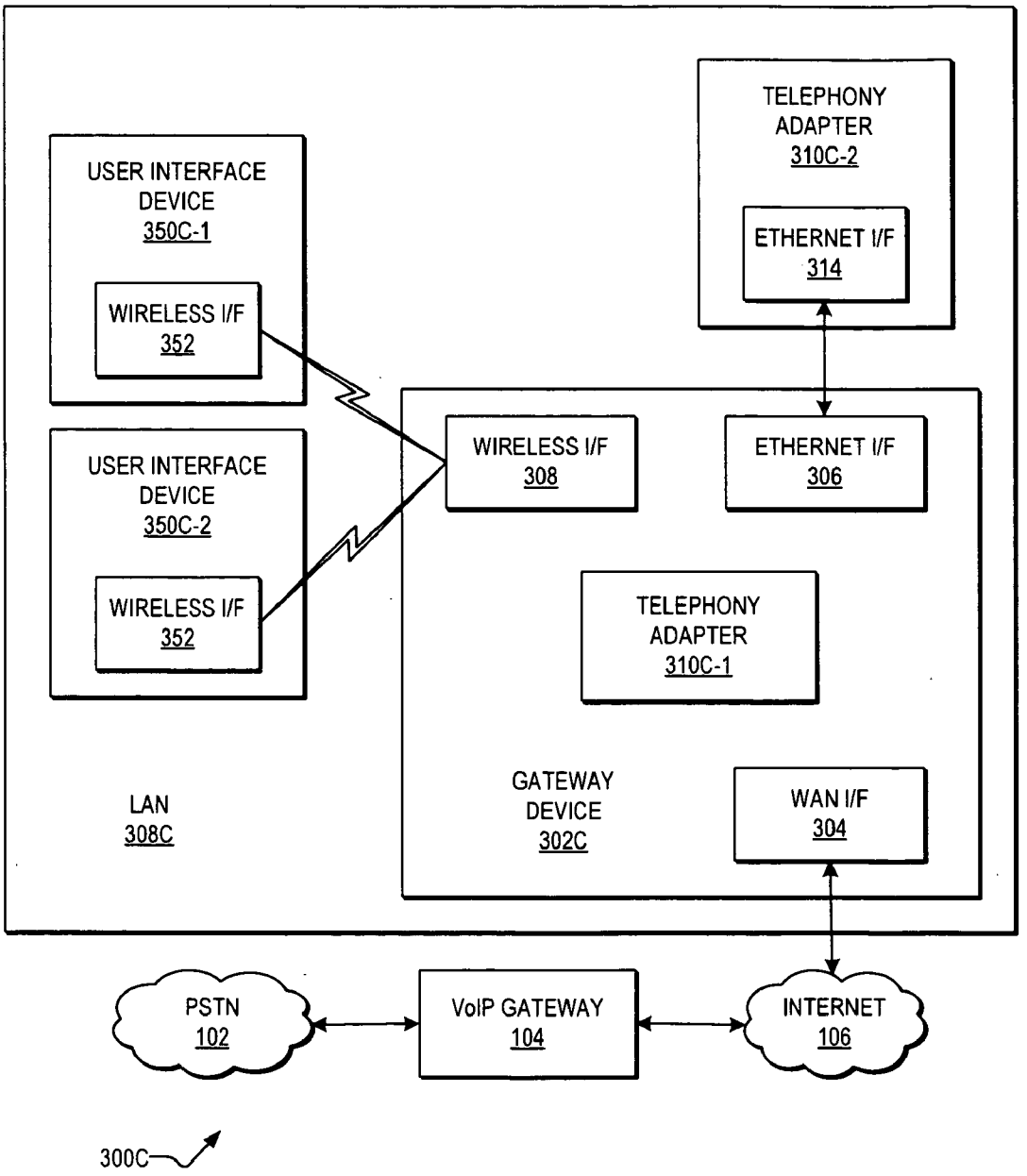

TELEPHONY ADAPTER 310C-2
ETHERNET I/F 314
USER INTERFACE DEVICE 350C-1
WIRELESS I/F 352
WIRELESS I/F 308
ETHERNET I/F 306
USER INTERFACE DEVICE 350C-2
WIRELESS I/F 352
TELEPHONY ADAPTER 310C-1
LAN 308C
GATEWAY DEVICE 302C
WAN I/F 304
PSTN 102
VoIP GATEWAY 104
INTERNET 106
300C

FIG. 3C

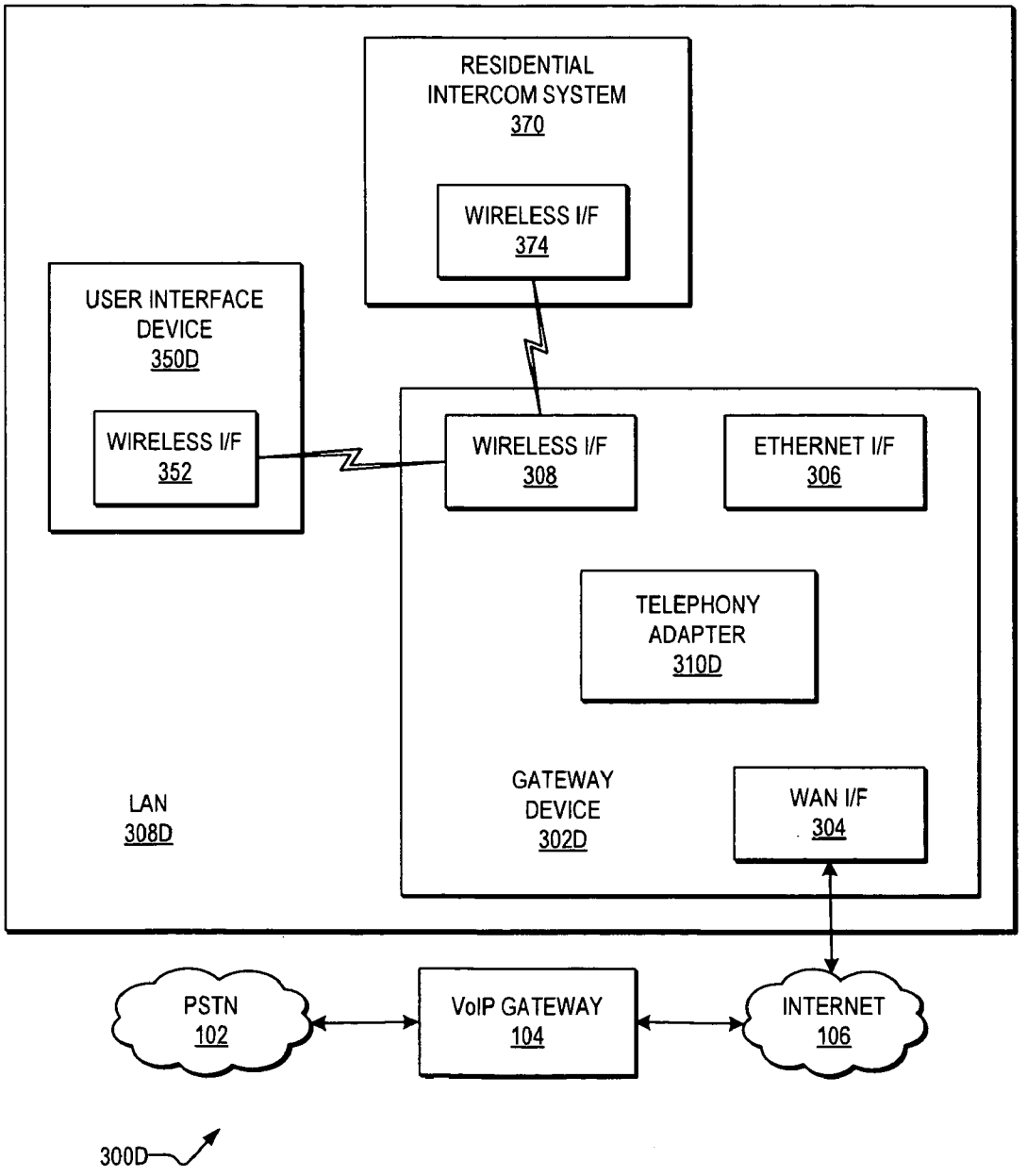

RESIDENTIAL
INTERCOM SYSTEM
370
WIRELESS I/F
374
USER INTERFACE
DEVICE
350D
WIRELESS I/F
352
WIRELESS I/F
308
ETHERNET I/F
306
TELEPHONY
ADAPTER
310D
LAN
308D
GATEWAY
DEVICE
302D
WAN I/F
304
PSTN
102
VoIP GATEWAY
104
INTERNET
106
300D

FIG. 3D

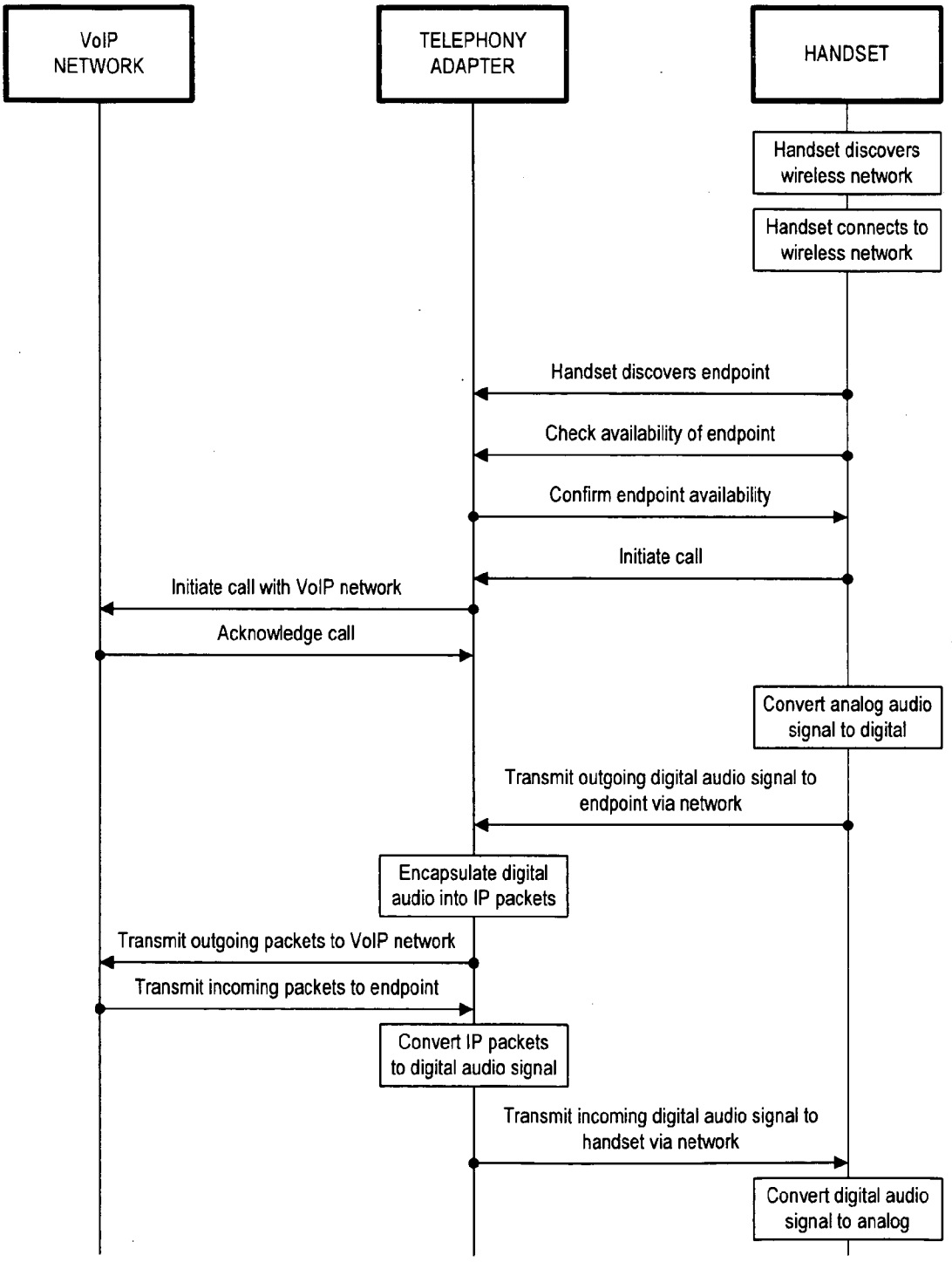

VoIP NETWORK
TELEPHONY ADAPTER
HANDSET
Handset discovers wireless network
Handset connects to wireless network
Handset discovers endpoint
Check availability of endpoint
Confirm endpoint availability
Initiate call
Initiate call with VoIP network
Acknowledge call
Convert analog audio signal to digital
Transmit outgoing digital audio signal to endpoint via network
Encapsulate digital audio into IP packets
Transmit outgoing packets to VoIP network
Transmit incoming packets to endpoint
Convert IP packets to digital audio signal
Transmit incoming digital audio signal to handset via network
Convert digital audio signal to analog

FIG. 4

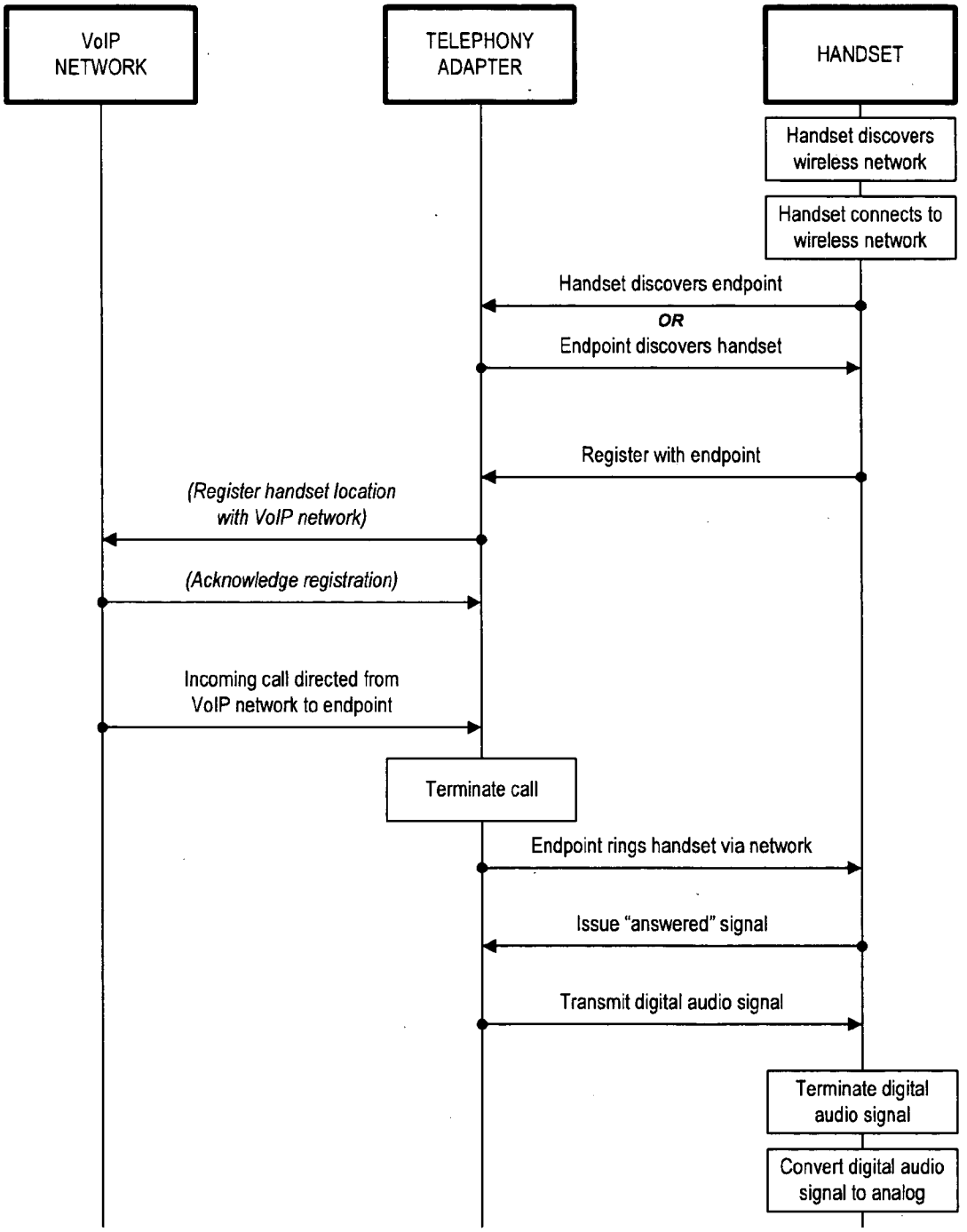

VoIP NETWORK
TELEPHONY ADAPTER
HANDSET
Handset discovers wireless network
Handset connects to wireless network
Handset discovers endpoint
OR
Endpoint discovers handset
Register with endpoint
(Register handset location with VoIP network)
(Acknowledge registration)
Incoming call directed from VoIP network to endpoint
Terminate call
Endpoint rings handset via network
Issue "answered" signal
Transmit digital audio signal
Terminate digital audio signal
Convert digital audio signal to analog

FIG. 5

DISTRIBUTED CODEC FOR PACKET-BASED COMMUNICATIONS

BACKGROUND

[0001] With the increase in residential broadband connections, the use of packet-based systems for voice communications has increased as well. One popular communications system is known as Voice over Internet Protocol (VoIP), which enables the routing of voice conversations over the Internet or other Internet Protocol (IP) networks.

[0002] One conventional VoIP implementation utilizes an analog telephone adapter (ATA), which is a device used to connect one or more standard analog telephones to a VoIP network. The ATA typically includes an Ethernet port for connection with a local area network (LAN) and a RJ-111 telephone port, into which users can plug a standard analog telephone. ATA devices typically use the SIP protocol, which is a standard for initiating, modifying, and terminating an interactive user session involving multimedia elements such as video, voice, and instant messaging. The ATA may be implemented as a standalone, dedicated ATA device, which can be connected to the Ethernet port on a router, or may be incorporated into a multi-function device, such as a residential gateway device, which includes a modem, router, and ATA. Each ATA is configured to support a predefined number of VoIP endpoints (typically one or two endpoints).

[0003] One disadvantage of conventional VoIP implementations is that all telephone devices (such as conventional analog telephones or fax machines) must be directly connected to the ATA. It is possible to connect multiple telephone devices to a single ATA by providing the ATA with multiple RJ-111 ports, or by connecting the ATA to a conventional cordless telephone base station having multiple handsets. However, a single ATA can only be used to terminate a predetermined limited number of telephone calls. Thus, if two analog telephones are connected to an ATA that only supports a single VoIP endpoint, only one of the telephones can be used at any given moment. Even if other ATAs are available on the same LAN and are not being used, the telephones connected to the first ATA cannot be used.

[0004] In other ATA implementations, the ATA may be combined with a telephone handset to provide a single integrated telephony device. Thus, the handset can have its own dedicated VoIP endpoint, network interface (such as, e.g., an Ethernet port or a wireless IEEE 802.11 interface), and audio subsystem (e.g., microphone and speaker). If multiple handsets are provided in a LAN, each handset is capable of terminating VoIP calls, thereby making all of the handsets simultaneously available to make telephone calls. A disadvantage of this arrangement is that the ATA functionality requires a significant amount of processing power to perform the necessary CODEC (coder-decoder) processes to digitize and compress the audio signal, and to encapsulate that signal into IP packets. This can dramatically increase the cost of each handset. In addition, the high power consumption of the ATA circuitry limits the amount of battery life available for wireless handset implementations.

[0005] Accordingly, there is a need for an improved packet-based communications system that enables ATA services to be more efficiently utilized in a network.

DESCRIPTION OF THE DRAWINGS

[0006] FIG. **1** is a block diagram showing an exemplary voice communications network environment **100** utilizing conventional VoIP implementations.

[0007] FIG. **2** is a block diagram showing an exemplary voice communications network environment, in accordance with embodiments of the present invention.

[0008] FIGS. **3**A-**3**D are block diagrams of various network environments, in accordance with embodiments of the present invention.

[0009] FIG. **4** is exemplary process flow for a telephone call initiated by a user interface device, in accordance with embodiments of the present invention.

[0010] FIG. **5** is exemplary process flow for an incoming telephone call received by the user interface device, in accordance with embodiments of the present invention.

DETAILED DESCRIPTION

[0011] In the following description, reference is made to the accompanying drawings which illustrate several embodiments of the present invention. It is understood that other embodiments may be utilized and mechanical, compositional, structural, electrical, and operational changes may be made without departing from the spirit and scope of the present disclosure. The following detailed description is not to be taken in a limiting sense, and the scope of the embodiments of the present invention is defined only by the claims of the issued patent.

[0012] Some portions of the detailed description which follows are presented in terms of procedures, steps, logic blocks, processing, and other symbolic representations of operations on data bits that can be performed on computer memory. Each step may be performed by hardware, software, firmware, or combinations thereof.

[0013] FIG. **1** is a block diagram showing a VoIP network environment **100** utilizing conventional VoIP implementations. Telephone calls with telephone numbers in the public switched telephone network (PSTN) **102** pass through a VoIP gateway **104**, which provides translation services between the PSTN **102** and endpoints coupled to IP networks, such as the Internet **106**. A wireless router/modem **108** serves as a gateway between a LAN **114** and the Internet **106**. The LAN **114** includes three ATAs **110***a*-**110***c*. The first ATA **110***a* comprises a standalone, dedicated ATA **110***a* having an Ethernet port for direct connection to the router **108** and a telephone port for direct connection to a conventional analog telephone **120***a*. The second ATA **110***b* comprises a dedicated ATA similar to the first ATA **110***a*. However, instead of being connected to a single telephone **120***a*, the second ATA **110***b* is connected to a standard analog cordless telephone base station **111**. Three handsets **120***b*-**120***d* share the single telephone line provided by the telephone base station **111**. The third ATA **110***c* is integrated into a WiFi handset **120***e*, which communicates wirelessly with the router **108**.

[0014] In this environment **100**, there are five devices **120***a*-**120***e* which could potentially be utilized to place telephone calls, but only three ATAs **110***a*-**110***c*, with each ATA **110***a*-**110***c* being configured to support a single VoIP endpoint. Because each device **120***a*-**120***e* is only capable of

operating with the ATA **110***a*-**110***c* to which the device **120***a*-**120***e* is coupled, in some situations a user may not be capable of placing a telephone call, even if there are ATAs available on the LAN **114**. For example, when a first user places a telephone call using the handset **120***b*, the second ATA **110***b* will be used by the handset **120***b* to terminate the VoIP call. If a second user wishes to place a telephone call using handset **120***c*, the second ATA **110***b* will not be available and the second user must wait until the first user is done, even though the first ATA **110***a* and third ATA **110***c* are unused and available.

[**0015**] In accordance with embodiments of the present invention, a user interface communications device, such as a telephone handset, can be used with a plurality of VoIP endpoints without requiring a direct connection between the communications device and the VoIP endpoint. This can enable multiple users, each having their own user interface device, to more efficiently share a limited number of VoIP endpoints available on a network. As a result, users can more efficiently utilize all of the VoIP endpoints existing on the network.

[**0016**] FIG. **2** is a block diagram showing an exemplary voice communications network environment **200**, in accordance with embodiments of the present invention. The network environment **200** comprises a user interface device **250** coupled to a telephony adapter **210** via a network. The network may comprise a wired or wireless network. The telephony adapter **210** is coupled to an IP network, such as the Internet **106**, via a router/modem **208**. A VoIP gateway **104** provides translation services between the protocols of the VoIP telephony adapter **210** and the PSTN **102**. The user interface device **250** provides the interface to the end user, and the telephony adapter **210** provides the VoIP endpoint functionality for communication with the VoIP gateway **104** and the interface to the user interface device **250**, as will be described in greater detail below.

[**0017**] In operation, one or more user interface devices **250** may utilize the telephony adapter **210** as a VoIP endpoint to terminate VoIP calls. The user interface device **25**Q, may be implemented as, e.g., a slim client handset, and is configured to serve as the audio subsystem by providing audio output using a speaker **272** and audio input using a microphone **274**. The IP signaling stack and CODEC functions, which are computationally intense, can be implemented on the telephony adapter **210**, thus relieving the user interface device **250** of much of the processing load. The primary functionality of the user interface device **250** may then be limited to merely managing communication with the telephony adapter **210**, and providing the user with bidirectional audio and key press functionality.

[**0018**] The physical and logical decoupling of the user interface device **250** from the VoIP endpoint enables a slim client handset to utilize any unused VoIP endpoint available on the network, even if that endpoint is not physically nearest to the handset. The user interface device **250** may operate essentially as a network attached portable streaming audio device. The user interface device **250** may provide the user with an analog input/output (e.g., microphone/speaker) and limited CODEC functionality, e.g., encoding of analog audio to digital audio (A/D conversion), and decoding of digital audio to analog audio (D/A conversion). Because telephone calls involve bidirectional audio streams, the user interface device **250** may appear as a digital audio source/sink to the telephony adapter **210**, and the telephony adapter **210** may appear to the user interface device **250** as a digital audio source/sink as well.

[**0019**] In the embodiment illustrated in FIG. **2**, the telephony adapter **210** is configured to advertise itself as a network service on the LAN and/or be discoverable by the user interface device **250**. The advertising and discovery process may be performed using a variety of protocols, such as, e.g., SLP (Service Location Protocol), SSDP (Simple Service Discovery Protocol), UPnP (Universal Plug and Play), SIP (Session Initialization Protocol), ZeroConf, and Bluetooth.

[**0020**] The telephony adapter **210** comprises a network interface **212** for coupling with a router **208** and may optionally comprise a second network interface **214** for coupling with a corresponding network interface **252** of the user interface device **250**. The second network interface **214** and the network interface **252** of the user interface device **250** may be coupled using, e.g., a wireless interface, such as IEEE 802.11 or Bluetooth.

[**0021**] The user interface device **250** further comprises an endpoint communications module **260**, an audio processing module **270**, and a user interface **256**, which can include, for example, a telephone keypad. The telephony adapter **210** further comprises a network protocol module **220**, a telephony signaling module **230**, a voice packet module **232**, and a client communications module **240**.

[**0022**] The audio processing module **270** of the user interface device **250** comprises an analog-to-digital converter (ADC) for converting analog audio signals from the microphone **274** into a digital audio signal and a digital-to-analog converter (DAC) for converting a digital audio signal into an analog audio signal to be output by the speaker **272**. In some embodiments, the user interface device **250** may be configured to display graphics or video, in which case the audio processing module **270** is further configured to process graphics and/or video signals. The audio processing module **270** may perform some compression/decompression as part of the codec functionality provided by the user interface device **250**.

[**0023**] The endpoint communications module **260** provides the functionality for communicating with the telephony adapter **210**. The endpoint communications module **260** provides the network connectivity functionality and the VoIP service discovery functionality.

[**0024**] In the telephony adapter **210**, the client communications module **240** is the counterpart to the endpoint communications **260** and enables communication between the user interface device **250** and the telephone adapter **210**. The client communications module **240** also provides the VoIP service advertisement functionality. The client communications module **240** and the endpoint communications module **260** may communicate using a variety of communication protocols for setting up the control and audio between devices, such as, e.g., SMS (Short Messaging Service) or SIP (Session Initialization Protocol). The bidirectional audio path can be over a packet-based network, such as IP, using, e.g., HTTP, TCP, RTP, or UDP.

[**0025**] The client communications module **240** receives digital audio signals from the user interface device **250** and

passes these signals to the voice packet module **232**, which processes the audio signal using voice CODECs and encapsulates the processed information into packets for transmission over an IP network, such as the Internet **106**. The audio processing performed by the voice packet module **232** may also include echo cancellation, voice compression, voice-activity detection, jitter removal, and clock synchronization. Similarly, the voice packet module **232** receives IP packets from the Internet **106** and converts them into digital audio signals to be transmitted to the user interface device **250**. In other embodiments, the processed information can be transmitted over different types of networks and need not be limited to only IP.

[**0026**] The telephony signaling module **230** detects call control/status information from the user interface device **250**, such as key presses into the user interface keypad **256**. The telephony signaling module **230** also collects destination address information needed to route telephone calls from the user interface device **250** to their intended destinations.

[**0027**] The network protocol module **220** implements the VoIP stack by receiving the packets from the voice packet module **232** and signaling output from the telephony signaling module **230**, and establishes the call and connection, and transmits the packets over the network (e.g., Internet **106**).

[**0028**] The user interface device **250** may be implemented in a variety of forms. For example, the user interface device **250** may comprise a slim client handset providing dedicated VoIP functionality. In other embodiments, the user interface device **250** may be implemented as a functional module of a multi-function device, such as a PDA (personal digital assistant) or a PC (personal computer). The telephony adapter **210** may be implemented, e.g., as a stand-alone dedicated VoIP device, or may be implemented as part of a multi-function device, such as a router or a PC.

[**0029**] FIG. **3**A is a block diagram of a network environment **300**A, in accordance with embodiments of the present invention. In this embodiment, the telephony adapter **310**A is implemented as part of a wireless gateway device **302**A, which includes a WAN (wide area network) interface **304** for coupling with the Internet **106**. The gateway device **302**A may provide both modem and router functionality, in addition to implementing the telephony adapter **310**A. The user interface device **350**A includes a wireless interface **352**, which provides wireless network connectivity to the wireless interface **308** of the gateway device **302**A using, e.g., IEEE 802.11. Thus, the user interface device **350**A may be located anywhere within wireless range of the gateway device **302**A.

[**0030**] FIG. **4** is exemplary process flow for a telephone call initiated by a user interface device **350**A (e.g., wireless a telephone handset), in accordance with embodiments of the present invention. First, the user interface device **350**A discovers and connects to a network (e.g., a wireless WiFi network). Next, a wireless connection between the endpoint communications module and the client communications module is established. This can be accomplished by the user interface device **350**A discovering the telephony adapter **310**A or vice versa. The user interface device **350**A may be configured to automatically discover all telephony adapters **310**A available on the network, and/or the telephony adapter **310**A may be configured to automatically discover all available user interface devices **350**A on the network Next, the user interface device **350**A determines whether the telephony adapter **310**A is available to serve as an endpoint for a new VoIP call (e.g., the user interface device **350**A determines whether the telephony adapter **310**A is idle or busy). If the telephony adapter **310**A confirms its availability, the user interface device **350**A initiates the call by entering the target telephone number using the numeric keys on the user interface **256**. These key presses are received by the telephony signaling module **230**, which locates the destination address information needed to route the telephone call through the WAN interface **304** and the Internet **106** to the VoIP gateway **104**, which acknowledges the call. The call is then routed via the PSTN **102** to a standard analog telephone number. Alternatively, the call may not utilize the PSTN **102** and may terminate at another VoIP device on the Internet **106**. If the telephony adapter **310**A is busy, the user interface device **350**A may proceed with checking the availability of another telephony adapter on the network.

[**0031**] At the user interface device **350**A, the user's voice is detected by (i.e., input to) the microphone **274** and converted into a digital audio signal by the audio processing module **270**. This digital audio signal is transmitted to the telephony adapter **310**A via the wireless network link between the wireless interface **352** and the wireless interface **308**. The voice packet module **272** encapsulates the digital audio signal into IP packets, and then transmits those IP packets to the VoIP gateway **104** via the IP network.

[**0032**] Incoming IP packets from the VoIP gateway **104** are received by the telephony adapter **310**A and converted into a digital audio signal. The IP packets correspond to the voice of the person on the other end of the telephone call. This digital audio signal is then transmitted by the client communications module **240** via the wireless link to the user interface device **350**A. This digital audio signal is then converted by the audio processing module **270** into an analog audio signal and output by the speaker (i.e. to the user's ear) **272**.

[**0033**] FIG. **5** is exemplary process flow for an incoming telephone call received by the user interface device **350**A, in accordance with embodiments of the present invention. First, the user interface device **350**A discovers and connects to a wireless network. Next, a wireless connection between the endpoint communications module **260** and the client communications module **240** is established. This can be accomplished by the user interface device **350**A discovering the telephony adapter **310**A or vice versa. Next, the user interface device **350**A registers its location with the telephony adapter **310**A to indicate that the user interface device **350**A is available to receive incoming telephone calls.

[**0034**] This registration can be performed in a variety of ways. For example, the user interface device **350**A may advertise its presence on the LAN **308**A, or the telephony adapter **310**A may periodically poll the LAN **308**A for the presence of user interface devices **350**. Once the user interface device is locally registered, the telephony adapter **310**A may serve as a proxy for handset registration so that when an incoming call is directed to the user interface device **350**A, this call is terminated by the telephony adapter **310**A. This registration process may be implemented in a fashion similar to the process by which a cellular telephone base

station informs a cellular network that a mobile handset is locally associated with that base station.

[0035] The telephony adapter **310**A is registered as an endpoint with the VoIP gateway **104**. In some embodiments, the telephony adapter **310**A may also register with the VoIP gateway **104** to indicate that the user interface device **350**A is associated with that telephony adapter **310**A. Thus, the telephony adapter **310**A indicates to the VoIP gateway **104** that all calls directed to the user interface device **350**A should be directed to the telephony adapter **310**A as the endpoint. The telephony adapter **310**A will then forward incoming calls to the user interface device **350**A. In other embodiments, incoming calls from the VoIP gateway **104** are directed to the telephony adapter **310**A, without any preregistration of the various user interface devices available on the local network. Once the incoming call is received and terminated by the telephony adapter **310**A, the appropriate user interface device is located on the local network and the A/V signal is forwarded to that user interface device.

[0036] After the telephony adapter **210** terminates the call, the telephony signaling module **230** issues a signal to the user interface device **250**A, causing the user interface device **250**A to ring. When a user answers the phone (e.g., by picking up the handset or pressing a button on the user interface **256**), an answer signal is transmitted back to the telephony signaling module **230**, indicating that the call has been accepted. Incoming audio signals from the IP network are depacketized by the telephony adapter **310**A and transmitted to the user interface device **350**A. The audio processing module **270** converts the digital audio signal into an analog signal and outputs it using the speaker **272**.

[0037] FIG. **3**B is a block diagram of a network environment **300**B, in accordance with other embodiments of the present invention. In this embodiment, the telephony adapter **310**B is a separate physical device from the gateway device **302**B and includes an Ethernet interface **314** for establishing network connectivity to the gateway device **302**B. The telephony adapter **310**B may be implemented as, e.g., a dedicated standalone telephony adapter, or as part of a multi-function device, such as a PC. The user interface device **350**B is similar to the user interface device **350**A, shown in FIG. **3**A, and establishes a network connection to the LAN **308**B via a wireless interface **352**.

[0038] When a user wishes to place a telephone call, the user interface device **350**B discovers the presence of the VoIP service on the LAN **308**B provided by the telephony adapter **310**B. Even though the user interface device **350**B does not have a direct connection to the telephony adapter **310**B, the user interface device **350**B can connect to the telephony adapter **310**B via the LAN **308**B. The bidirectional digital audio signals and call control signals can be routed by the gateway device **302**B between the two devices **350**B, **310**B, and the telephony adapter **310**B can perform all of the VoIP call routing and signal processing functions.

[0039] FIG. **3**C is a block diagram of a network environment **300**C, in accordance with other embodiments of the present invention. In this embodiment, a first telephony adapter **310**C-**1** is provided as part of the gateway device **302**C. A second telephony adapter **310**C-**2** is also provided on the LAN **308**C as a separate device coupled to the gateway device **302**C via an Ethernet interface **314**. Two user interface devices **350**C-**1** and **350**C-**2** are wirelessly connected to the gateway device **302**C via wireless interfaces **352**.

[0040] When a user at the first user interface device **350**C-**1** wishes to place a telephone call, the first user interface device **350**C-**1** discovers the presence of VoIP services on the LAN **308**C provided by the telephony adapters **310**C-**1** and **310**C-**2**. When multiple telephony adapters **310** are available on the LAN **308**C, the user interface device **350**C-**1** may be configured to default to a preferred telephony adapter **310**, or may be configured to automatically select from the multiple adapters (e.g., by connecting to the first adapter discovered, to the first idle adapter discovered, or to the adapter in closest physical proximity). The call from the first user interface device **350**C-**1** can then be completed as described above.

[0041] If a second user at the second user interface device **350**C-**2** wishes to place a telephone call, the second user interface device **350**C-**2** also attempts to discover the presence of VoIP services on the LAN **308**C. Because the first telephony adapter **310**C-**1** is already being utilized to terminate the VoIP call for the first user interface device **350**C-**1**, it is not available to terminate the call for the second user interface device **350**C-**2**. The second user interface device **350**C-**2** can then search the LAN **308**C for other available telephony adapters **310**. Alternatively, the second user interface device **350**C-**2** may automatically discover the presence and status (e.g., idle or busy) of every telephony adapter on the network during the initial discovery process. When the second user interface device **350**C-**2** discovers the second telephony adapter **310**C-**2**, the user interface device **350**C-**2** can initiate the VoIP call, as described above with respect to FIG. **3**B. For this second telephone call, the gateway device **302**C is utilized by the user interface device **350**C-**2** to route the call control and bidirectional audio signals between the user interface device **350**C-**2** and the telephony adapter **350**C-**2**; the first telephony adapter **310**C-**1** is not used.

[0042] An advantage of this arrangement is that all of the available telephony adapters **310** on the LAN **308**C may be more efficiently utilized. The user interface devices **350** can utilize any of the available telephony adapters **310** to initiate a VoIP call, and the telephony adapters **310** can utilize any of the available user interface devices **350** so that a VoIP call can be answered.

[0043] In accordance with other embodiments of the present invention, audio and/or video from a telephony adapter may be received by a user interface device that lacks a user input or a device having a user input that is not used for receiving telephone calls (e.g., keypad or audio microphone). For example, a networked television, DMA (digital media adapter), or stereo may be configured to play audio terminated by the telephony adapter.

[0044] FIG. **3**D is a block diagram of a network environment **300**D, in accordance with other embodiments of the present invention. In this embodiment, a user interface device may be implemented in a device lacking a user input (e.g., a residential intercom system **370**). The telephony adapter **310**D terminates an incoming VoIP telephone call and directs the audio signal to the residential intercom system **370** via a wireless interface **374**. Thus, an incoming VoIP audio stream can be sent to a device not traditionally

capable of playing live telephone calls. In one example, a user may place a telephone call to a phone number associated with some device located in the user's house. The telephony adapter **310**D terminates the VoIP call and outputs the audio signal (e.g., the user's voice) over the target device, e.g., an intercom system, television, stereo, DMA, or other networked device.

[**0045**] In some embodiments, the telephony adapter **310**D may be configured to direct the calls to alternate devices if the primary target device is unavailable. For example, if an incoming telephone call is directed to the telephone handset user interface device **350**D, and the user interface device **350**D is unavailable (e.g., if the device **350**D is powered down), then the telephony adapter **310**D may direct an audio stream to the intercom system **370**, television, or other nearby device. The audio stream may include a prerecorded message instructing the user to turn on the user interface device **350**D, or may be the incoming audio signal from the telephone call. The telephony adapter **310**D may also be configured to transmit the audio signal to a series of fallback alternate devices, in the event that the intercom system **370** is unavailable. The series of alternate devices may be configured by the user in advance, or may be selectable during the call by the caller. For example, the telephony adapter **310**D may terminate the call and present the caller with an audio menu of target devices for connection. Thus, the caller may select "1" to output on the living room television, or "2" to output in the stereo in the kitchen.

[**0046**] In yet other embodiments, the incoming signal may be directed to a first user interface device and the outgoing signal may originate from a second user interface device. For example, an incoming audio and/or video call may be terminated by a telephony adapter, which transmits the incoming audio/video signal to a first user interface device having only output capability (e.g., a television or stereo). The receiving user may then utilize a second user interface device having only input capability (e.g., a wireless microphone, digital camera, or webcam) in order to respond to the call. Because the call is terminated by the telephony adapter, the division of functionality between multiple user interface devices is transparent to the caller.

[**0047**] In accordance with other embodiments of the present invention, systems may be provided for other types of session-based audio/video transmissions to slim client devices. For example, an advertising device in a retail location may be provided with a telephony adapter for detecting the presence of slim clients. The telephony adapter may then push advertisements to the passing users by effectively "calling" the user interface devices, as described in the embodiments above, and initiating an audio-visual data transfer. These slim clients can be, e.g., PDAs (personal digital assistants), telephone handsets, DMAs, or other devices capable of receiving an audio/video stream that have been preregistered to receive advertisements. The advertisements may include text, graphics, audio, and/or video and may comprise, e.g., in-store coupons, music clips (for a music store), audio book clips (for a bookstore), or movie trailers (for a video store).

[**0048**] In yet other embodiments, two user interface devices may communicate directly with each other. Thus, the applications may expand beyond the server/client relationship between the telephony adapter and the user interface device, and allow client/client networks between individual users. Thus, individual portable devices may share and/or trade information (e.g., coupons or advertisements) simply by proximity.

[**0049**] In accordance with some embodiments, security and privacy concerns may be addressed by allowing user interface devices to selectively enable or disable the receiving of calls from certain types of telephony adapters. For example, a user may opt out of all commercial advertisements by configuring the user interface device to block all such calls.

[**0050**] Embodiments of the present invention may be implemented in any of a variety of network topologies. For example, in a mesh network architecture, the VoIP endpoints may be co-located with mesh routers, and handsets are implemented as mesh endpoints. Thus, the mesh network can locate the path from a handset to the closest available VoIP endpoint.

[**0051**] In accordance with some embodiments, the handset may store a small amount of personalization information for the user, while the bulk of the user's personal data is stored elsewhere on the network to be retrieved by the user interface device and/or the telephony adapter when needed. For example, the user identity (e.g., telephone number or login name) may be stored on the slim client handset, while contact lists may be maintained in a central server.

[**0052**] Embodiments of the present invention may provide various advantages not provided by prior art systems. In particular, by implementing the VoIP signaling stack and CODECs in the telephony adapter **210**, the user interface device **250** may be implemented in low cost fashion using any device that provides a network connection and bidirectional audio capability. Alternatively, the user interface device **250** could implement an advanced CODEC that is not used by the telephony adapter **210**. This would allow the telephony adapter **210** to use the bidirectional audio stream as is and possibly provide an improved user experience.

[**0053**] In addition, the technical challenges typically associated with managing locally roaming voice handsets may be significantly mitigated. The telephony adapter implements the VoIP endpoint, not the user interface device. The user interface device functions as an extension of the audio media path which is established or terminated at the VoIP endpoint in the telephony adapter. Thus, the concept of "local" roaming (i.e., on a single premises) is effectively equivalent to momentarily unplugging an audio I/O device (e.g., a microphone/speaker headset) from one socket and plugging it into another socket. In other words, because the telephone call is terminated at the telephony adapter, the continuous connection between the user interface device and the telephony adapter need not be required in order to maintain the telephone call. Thus, the call can maintain a fixed termination point at the telephony adapter, while the audio path from the telephony adapter to the user interface device may be interrupted or change between multiple sets of microphones and speakers. Furthermore, the bidirectional audio path between the telephony adapter and the user interface device would not be required to use the same audio CODEC in each direction. For example, the user interface device and the destination device may both be provided with CODECs for processing stereo quality audio. However, the telephony adapter may only be provided with CODECs for

processing telephone quality audio. The signal from the user interface device may still be processed by the telephony adapter and routed over the Internet.

[0054] In addition, because the user interface device and the telephony adapter are connected over a LAN, the connection may be temporary and configurable, thus enabling improved portability and flexibility. In contrast, conventional cordless phones are associated with a particular base station and communicate over a fixed RF signal, rather than a network.

[0055] In addition, systems in accordance with the present invention may provide VoIP functionality for multiple user interface devices without centralized VoIP signaling or voice coding. Thus, these systems may be conceptualized a distributed private branch exchange (PBX).

[0056] While the invention has been described in terms of particular embodiments and illustrative figures, those of ordinary skill in the art will recognize that the invention is not limited to the embodiments or figures described. For instance, some of the examples provided above describe the use of one or more telephony adapters in a home or small office setting. In other embodiments, the telephony adapters may be implemented in a business or public setting. For example, a public venue may host a WiFi hotspot including plurality of VoIP telephony adapters, which would advertise themselves as a VoIP service to users of the hotspot. A guest or visitor to that public venue could activate their handset, discover the service (e.g., by actively inquiring for the service), and place a telephone call using any available telephony adapter. In other embodiments, the handset may subscribe to a service that enables the handset to passively discover the existence of all available telephony adapters by listing for their advertisements. After the handset has been activated and registered by a telephony adapter, the telephony adapter can relate the location of the handset to the VoIP network, thereby enabling subsequent calls directed to that handset to be forwarded via that telephony adapter.

[0057] The program logic described indicates certain events occurring in a certain order. Those of ordinary skill in the art will recognize that the ordering of certain programming steps or program flow may be modified without affecting the overall operation performed by the preferred embodiment logic, and such modifications are in accordance with the various embodiments of the invention. Additionally, certain of the steps may be performed concurrently in a parallel process when possible, as well as performed sequentially as described above.

[0058] Therefore, it should be understood that the invention can be practiced with modification and alteration within the spirit and scope of the appended claims. The description is not intended to be exhaustive or to limit the invention to the precise form disclosed. It should be understood that the invention can be practiced with modification and alteration and that the invention be limited only by the claims and the equivalents thereof.

What is claimed is:

1. A packet-based communications system, comprising:

a client communications module for transmitting digital audio signals to a user interface device over a local area network (LAN) and for receiving digital audio signals from the user interface device;

a voice packet module configured to:

receive audio signals from the client communications module and encapsulate the audio signals into data packets; and

convert data packets into audio signals and transmit the audio signals to the client communications module; and

a network protocol module for transmitting data packets between the voice packet module and an Internet Protocol (IP) network.

2. The system of claim 1, wherein:

the client communications module is configured to register as a service for the LAN.

3. The system of claim 1, wherein:

the client communications module is configured to direct telephone calls from the network protocol module directed to any of a plurality of user interface devices and, in response to receiving a telephone call from the network protocol module for an identified user interface device, to locate the identified user interface device on the LAN.

4. The system of claim 3, wherein:

the client communications module is further configured to direct audio signals associated with the telephone call to the identified user interface device.

5. The system of claim 1, further comprising:

a telephony signaling module coupled for receiving call control information from the client communications module and the network protocol module.

6. The system of claim 1, wherein the client communications module is further configured to:

receive a telephone call from the network protocol module directed to an identified user interface device; and

if the identified user interface device is not available to receive the telephone call, to transmit an audio message to an audio device on the LAN.

7. The system of claim 1, wherein the client communications module is further configured to:

receive a registration from a user interface device; and

identify a location of the user interface device to a VoIP gateway.

8. The system of claim 1, further comprising:

a wireless interface for communicating with the user interface device.

9. The system of claim 1, wherein:

the voice packet module is further configured to:

convert data packets into video signals and transmit the video signals to the client communications module.

10. The system of claim 9, wherein:

the voice packet module is further configured to:

receive video signals from the client communications module and encapsulate the video signals into data packets.

11. A communications device, comprising:

a microphone;

a speaker;

a user input device;

a network interface for coupling the communications device to a local area network (LAN);

an audio processing module comprising an analog-to-digital converter (ADC) for converting analog audio signals from the microphone into a digital audio signal and a digital-to-analog converter (DAC) for converting a digital audio signal into an analog audio signal, said analog signal being output by the speaker; and

an endpoint communications module for connecting to a telephony adapter on the LAN via the network interface, said endpoint communications module communicating digital audio signals with the telephony adapter over the LAN.

12. The communications device of claim 11, wherein:

said endpoint communications module is further configured to locate a first telephony adapter on the LAN, and if the first telephony adapter is not available to complete a telephone call, to locate a second telephony adapter on the LAN and to communicate digital audio signals with the second telephony adapter over the LAN.

13. The communications device of claim 11, wherein:

said network interface comprises a wireless network interface.

14. The communications device of claim 11, further comprising:

a video display, wherein said endpoint communications module is configured to receive digital video signals from the telephony adapter over the LAN.

15. The communications device of claim 14, further comprising:

a video display, wherein said endpoint communications module is configured to receive digital video signals from the telephony adapter over the LAN.

16. A voice over IP (VoIP) communications system, comprising:

a VoIP module for terminating a VoIP call;

a client communications module for transmitting digital audio signals to a user interface device over a local area network (LAN) and for receiving digital audio signals from the user interface device; and

an audio processing module configured to:

receive audio signals from the client communications module and encapsulate the audio signals into data packets to be transmitted by the VoIP module; and

convert data packets from the VoIP module into audio signals and transmit the audio signals to the client communications module.

17. A packet-based communications method, comprising:

establishing a communications session over an Internet Protocol (IP) network between a telephony adapter and a target device;

receiving digital audio signals at the telephony adapter from a user interface device via a local area network (LAN);

encapsulating the audio signals into data packets; and

transmitting the data packets to the target device via the IP network.

18. The method of claim 17, wherein:

the communications session comprises a Voice over Internet Protocol (VoIP) telephone call.

19. The method of claim 17, further comprising:

receiving incoming data packets from the target device via the IP network;

converting the incoming data packets into digital audio signals; and

transmitting the digital audio signals to the user interface device.

20. The method of claim 17, further comprising:

receiving a VoIP telephone call from the IP network for an identified user interface device;

locating the identified user interface device on the LAN; and

terminating the VoIP telephone using the telephony adapter.

21. The method of claim 20, further comprising:

directing audio signals associated with the VoIP telephone call to the identified user interface device.

22. The method of claim 17, further comprising:

receiving a registration from the user interface device; and

identifying a location of the user interface device to a VoIP gateway.

23. The method of claim 17, wherein:

said receiving digital audio signals at the telephony adapter from the user interface device comprises receiving digital audio signals via a wireless interface.

24. A packet-based communications system, comprising:

a client communications module for transmitting digital audio signals to a user interface device over a local area network (LAN) and for receiving digital audio signals from the user interface device;

a means for receiving audio signals from the client communications module and encapsulate the audio signals into data packets; and

a means for converting data packets into audio signals and transmitting the audio signals to the client communications module; and

a means for transmitting data packets between the voice packet module and an Internet Protocol (IP) network; and

25. A communications device, comprising:

a microphone;

a speaker;

a user input device;

a means for interfacing with a local area network (LAN);

a means for audio processing comprising an analog-to-digital converter (ADC) for converting analog audio signals from the microphone into a digital audio signal and a digital-to-analog converter (DAC) for converting a digital audio signal into an analog audio signal, said analog signal being output by the speaker; and

a means for connecting to a telephony adapter on the LAN via the means for interfacing with a LAN and communicating digital audio signals with the telephony adapter over the LAN.

26. The communications device of claim 25, wherein:

said means for connecting and communicating is further configured to locate a first telephony adapter on the LAN, and if the first telephony adapter is not available to complete a telephone call, to locate a second telephony adapter on the LAN and to communicate digital audio signals with the second telephony adapter over the LAN.

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